



A METHOD TO DETERMINE THE TRANSFER CHARACTERISTIC
OF A MICROPHONE SYSTEM, AND MICROPHONE SYSTEM.

The present invention relates to a method defined in the preamble of claim 1 and to a
5 microphone system defined in claim 9.

When receiving and processing acoustic signals, there is frequently a need to design
microphone systems with a transfer characteristic such as to generate the electrical output
signal as a predetermined or predeterminable function of the angle of incidence of the acoustic
signals. In particular there is a need to design microphone systems with a predetermined or
10 predeterminable directional characteristic such that acoustic signals from certain directional
ranges shall be at a higher gain, from other zones at lesser ones, when transforming them into
the output signal, and this need extends to systems with a unidirectional receiving
characteristic.

Many procedures are known to implement such transfer characteristics. Illustratively
15 the state of the art comprises the patent documents WO99/04598, corresponding to US
09/146,784 (ϕ multiplication) or WO99/09786 corresponding to US 09/168,184 (ϕ filter control)
of this applicant, whereby, basically, desired microphone-system transfer characteristics are
obtained from the phase shifts of acoustic signals incident on said microphone systems and
by appropriately processing of said signals.

20 The objective of the present invention is to propose another method to implement a
desired transfer characteristic in the above-discussed sense.

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This problem is solved by the invention by a method of the initially cited kind wherein
the microphone system comprises at least two microphone sub-systems of which the transfer
characteristics differ in relation to said direction regarding the electric output signals of each,
25 and in that the output signal is formed as a mathematical product which is saturated at a
predetermined or predeterminable value, the ratio of the output signals from the said
microphone sub-assemblies being a factor in said product.

The expression "saturation" within the scope of the present invention denotes that the value of a mathematical function under consideration shall be clipped once it has reached a predetermined value and that as a result said value remains constant, contrary to the mathematical function per se.

Even though a low-value saturation of said product, that is of the weighted ratio, may be appropriate, preferably the product shall be saturated at a maximum value.

Moreover the second factor of the saturated product may assume an arbitrary value other than zero, hence also the value of 1.

In another preferred embodiment, the cited function comprises a difference between an adjustable constant and the saturated product, preferably the value of the constant being selected to be at least approximately equal to the saturation value.

Preferably again the cited ratio is obtained from the output-signals' amplitudes without regard to their phases.

In an especially preferred implementation of the method of the invention, the said ratio is used within the following function:

$$S = c_d [A - (\alpha \parallel \frac{|c_n|}{|c_d|}) \parallel_{satB}]$$

where

S is the output signal of the microphone system, A is predetermined or predeterminable signal value, $|c_d|$ is the amplitude of the output signal from a first sub microphone-system of which the transfer characteristic is at a maximum gain at one angle of incidence, the characteristic to be formed also being at maximum gain, $|c_n|$ is the output-signal amplitude of the second sub microphone-system, satB is the ratio saturation at a predetermined or predeterminable maximum signal value B, and α is a predeterminable or predetermined factor.

In an especially preferred implementation of the method of the invention applicable to hearing aids, the transfer characteristics of the sub microphone-systems are selected in such manner that they shall transmit, in substantially mutually opposite directions and at maximum gain, signals from incident acoustic inputs.

A microphone system of the invention and of the initially cited kind is characterized in that the processing unit includes a weighted-ratio forming unit fitted with a denominator input, a numerator input and a weighting input, the numerator and denominator inputs being operationally connected to the input of a processing unit, further the weighted-ratio forming unit which generates an output signal saturated at a maximum and/or a minimum at its output and which is operationally connected to the output of the processing unit.

Preferred embodiment variations of the microphone system of the invention are specified in claims 10 through 18.

The method as well as the microphone system of the invention are especially applicable to hearing aids.

Even though the method of the invention and the microphone system of the invention may easily be implemented in the manner of time-domain signal processing, signal processing in a preferred embodiment is carried out in the frequency domain using time-domain/frequency converters or frequency-domain/time-domain converters.

The invention is elucidated below in relation to the Figures of the drawing.

Figs. 1a, 1b illustrate the transfer characteristics of two sub microphone-systems "a" and "b" operated in the manner of the invention,

Fig. 2 shows the angle ϕ as a coordinate axis in relation to Figs. 1a, 1b and, in dB, further the ratio function Q based on the characteristics of Figs. 1a and 1b, and also the saturation of the ratio at the maximum value of 0 dB,

Fig. 3 is based on the saturated ratio of Fig. 2, also this saturated function as a linear gain scale and the formation of a function F from the difference between said saturated ratio and to a fixed value,

Fig. 4 is a view similar to Figs. 1a, 1b and shows, in shading, a transfer characteristic of the invention,

Fig. 5 is a view similar to Fig. 4 of another transfer characteristic implemented by the invention, and

Fig. 6 is a simplified signal-flow and functional block diagram of the implementation of a microphone system of the invention.

Without claiming scientific rigor, the method of the invention shall be represented in Figs. 1 through 3 by means of simple transfer characteristics, each a cardioid of first order. In the light of this simple method, the expert easily understands how, in the invention, and using more complex transfer functions, a desired transfer characteristic can be attained.

sub a 4 } A first sub-microphone system is designed with a three-dimensional transfer characteristic shown in two dimensions in Fig. 1a and relating to its transfer or gain features of acoustic signals incident on said system from the direction ϕ . Fig. 1b is similar to Fig. 1a of a transfer characteristic of a second sub-microphone system which is assumed mirror-symmetrical to the axis $\pi/2$; $3\pi/2$ of the transfer characteristic of the first sub-microphone system. The transfer characteristics of Figs. 1a and 1b resp. are denoted by c_d and c_d .

sub a 5 } In Fig. 2, the transfer functions c_d and c_d are shown qualitatively and in dB relative to the ϕ coordinate axis of Figs. 1a and 1b.

sub a 6 } As regards the acoustic unit signals incident on the two sub microphone-systems, the transfer characteristics shown in Figs. 1 and 1b simultaneously correspond to the signal values at the outputs of the sub microphone-systems under consideration.

sub a 7 } In the invention a ratio Q is formed from these two values of output signals, again denoted by c_d and c_d , for instance

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$$Q = |c_n| / |c_d|$$

sub a 9 } This ratio leads to the function Q shown qualitatively in dot-dash lines in Fig. 2 with a singularity at $\phi = \pi$. When the ratio is real, the singularity resulting at the null position of the

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denominator $|c_d|$ is anyway clipped, that is, the ratio function Q is saturated. Preferably the ratio is saturated at a predetermined or predeterminable value B , preferably as shown in Fig. 1 at the value "1" at the maximum value of the transfer functions of Figs. 1a, 1b of "1".

Be it assumed now that the denominator transfer characteristic -- in the present case c_d -- is one at which the desired transfer characteristic be the dominant one, namely a transfer characteristic with a high signal gain in a given angular range wherein the desired characteristic to be implemented also shall have high signal gain, then the advantage of forming the ratio of the invention becomes clear. Said transfer characteristic -- which is dominant for the desired result -- produces a singularity of the ratio in the angular range around zero. However the zero-point angular range of the dominant transfer characteristic, or of those angular ranges with reduced signal gains shall be those which must be altered, ie to be "improved" in order to attain the desired characteristic. It is precisely there that the possibility exists for a straightforward intervention, namely by saturating at a predetermined or predeterminable constant ratio value.

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For reasons of clarity, the saturated-ratio function Q_{sat1} is shown with a linear gain scale in Fig. 3 at 1. Fig. 3 further shows that in the unsaturated angular ranges, in the present case between 0 and $\frac{1}{2}\pi$ and between $3\pi/2$ and 2π , the saturated ratio Q_{sat1} is a directional transfer-characteristic function. If now specific directional characteristics are desired for the transfer characteristic, then the range of the ratio which was set in the invention to a predetermined saturation value, in this case to 1, shall be used to achieve therein, that is within this angular range, a defined minimum gain in the desired transfer characteristic. This goal is attained in the embodiment being discussed in that the saturated ratio is subtracted from a predetermined or predeterminable fixed value A , in the present illustration for instance and preferably having the value of 1. The result is a function F again shown as a full line in Fig. 3,

$$F = A - Q_{sat1}B$$

or, as a special and preferred case

$$F = 1 - Q_{\text{sat}1}$$

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cont It follows that a transfer function F was attained with a vanishing signal gain except in the

range $0 \leq \phi \leq \frac{1}{2} \pi$ and $3\pi/2 < \phi < \pi$.

The following explanations now can be offered relating to the method of the invention:

5 -- Basically the transfer characteristic to be attained is implemented at the output of the microphone system of the invention as a function of a ratio of the output signals from two microphone sub-systems of different transfer characteristics, where said ratio is saturated at a predetermined or predeterminable maximum value.

10 Preferably and elucidated further below, the ratio function Q is multiplied as one factor with a further predetermined or predeterminable fixed weighting factor before saturation is applied to the resulting mathematical product. Said weighting factor in the example shown in Figs. 1 through 3 is 1.

It may furthermore be highly advantageous to carry out the saturation on the product of said factor and the ratio, also when reaching predetermined minimum values.

15 -- The ratio may be formed directly by dividing the signal amplitudes, irrespective of phase.

sub-a 11 -- Even though the saturated product might be used in the form of another function, generally therefore as $F = F[(\alpha Q)\text{sat}B]$, far more preferably the implementation of a directional characteristic shall be by means of subtracting the said saturated product from
20 a predetermined or predeterminable fixed value.

As elucidated further below, varying the cited fixed value and/or the multiplicative factor α of the saturated product allows, in exceedingly simple manner, to vary the desired directional characteristic.

25 -- Basically the sub microphone-systems may be in the form of all known microphones and their combinations, which shall be designed for different transfer

characteristics as required by their operating positions and regarding the angle of incidence ϕ of acoustic signals.

-- Sub microphone-systems are preferentially used especially as regards attaining directional characteristics when their transfer characteristics are identical while being directionally mutually opposite as regards the angle of incidence of acoustic signals.

-- Such microphone systems can be implemented in particular using the known "delay and add" principle.

The above mentioned directionally mutually opposite operational microphone systems can be implemented in particular also when such a system involves two microphones of which the outputs -- in a manner shown below -- are each time-delayed and are correspondingly added in order to form the two microphone sub-systems.

-- It is understood that the method of the invention can be expanded using three or more sub microphone-systems in order to attain highly complex transfer functions and combinations of the latter.

In summary, the transfer function preferably used in the invention is shown again, namely

$$S = c_d \left[A - \left(\alpha \parallel \frac{|c_n|}{|c_d|} \right) \parallel_{satB} \right].$$

Fig. 4 shows the transfer function constituted by the inversely directional, identical cardioid transfer characteristics C_a of the invention, corresponding to the transfer function

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$$S' = c_d \left(1 - \left[1 - \frac{|c_n|}{|c_d|} \right]_{sat1} \right)$$

Fig. 5 shows the resultant transfer characteristic where applicable:

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$$S'' = c_d \left(1 - \left[4 \frac{|c_n|}{|c_d|} \right]_{sat1} \right)$$

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Fig. 6 illustratively shows a microphone system operating in the manner of the method of the invention by means of a simplified signal-flow functional block diagram and especially applicable also to hearing aids.

As shown in Fig. 6, the microphone system comprises at the input side a system 1 with at least two sub microphone-systems 1a and 1b. The output signals O_{1a} and O_{1b} at the outputs of said sub-systems are a function of the direction ϕ of the acoustic signals incident on the input-side microphones. As shown in Fig. 6, the two sub microphone-systems definitely may consist of a single pair of microphones of which the outputs are coupled to each other in the "delay-and-add" technique. What is essential is that basically the signals at the outputs O_{1a} and O_{1b} are of different transfer characteristics as regards the acoustic signals incident at an angle ϕ .

Preferably the output signals O_{1a} and O_{1b} are fed to time-domain/frequency-domain converter units FFT 3a and 3b resp. provided and, as preferred, the subsequent signal processing take place in the frequency domain. Said outputs are operationally connected to inputs I_{5a} and I_{5b} resp. of magnitude-forming units 5a and 5b. The outputs of said magnitude-forming units are, as represented in Fig. 6, fed to the numerator and denominator inputs N and D of a divider unit 7. The output signal O_7 is multiplied by a weighting unit 9 by a predeterminable weighting factor α present at the control input C_9 and is operationally connected to the input I_{11a} of a subtraction unit 11.

As shown in dashed lines in Fig. 6, the divider unit 7 and the weighting unit 9 constitute a weighted ratio-forming unit 10. The factor α which illustratively in Fig. 6 is shown adjustable at the weighting unit 9 may assume values arbitrarily different from 0.

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Fig. 6 furthermore diagrammatically shows the signal at the output O_9 of the weighted ratio-forming unit 10 being fed to a saturation unit 12 of which the output is first fed to the input I_{11a} . The output signal of the weighted ratio-forming unit 10 may be saturated downward at the saturation unit 12 -- which obviously may be integral with this weighted ratio-forming unit 10 -- (shown dashed in the block 12 of Fig. 6) and/or upward at a predetermined or predeterminable value B (as schematically indicated at the input "satB". Preferably this setting shall also be at a maximum value. The signal applied to the subtraction unit 11 is subtracted from the fixed value A which is set or can be adjusted at the second input I_{11b} . The output signal O_{11} of the subtraction unit 11 is operationally connected to the input I_{12a} of a multiplication unit 13 of which the second input I_{13b} receives the output signal of that microphone sub-system 1a which is also applied to the denominator input N of the divider unit 7. If it is desired to change the angular saturation range discussed in Figs. 1 through 3, then the denominator signal and where called for also the numerator signal, which are fed to the inputs D and N resp. of the divider input 7, may be weighted further.

The output signal S_{out} of the microphone system of the invention appears at the output of the multiplier 13. Said signal includes the desired transfer characteristic as a function of the solid angle ϕ at which acoustic signals impinge on the input of the microphone system 1.

As already mentioned, preferably the selected transfer characteristics of the microphone sub-systems 1a and 1b shall be identical but mutually directionally opposite characteristics. By adjusting the weighting factor α , the saturation value B, the fixed value A, and, where called for, further weighting factors such as β , the desired transfer characteristics shall have been adjusted at the output signal S_{out} .

The method of the invention and the microphone system of the invention are unusually appropriate for hearing aids, also on account of economical signal processing and, as shown by Figs. 3 and 4, the remarkable ability to suppress signal transmission from undesired directions of incidence, for instance to the rear of a hearing aid. As regards hearing aids,

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cont preferably the sub microphone-systems having cardioid characteristics C_a shall be replaced
with sub-systems having hypercardioid characteristics H_{ca} (Fig. 5).